NPL Search Results

13/5/1 (Item 1 from file: 8)
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Network transfer, digital signal processing and display of CSU CHILL digitized radar signals
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Editor(s): Jayasumana, A.P.; Chandrasekar, V.
Editor(s) Affil.: Colorado State University, United States
Conference Title: Technologies, Protocols, and Services for Next-Generation Internet
Conference Location: Denver, CO United States **Conference Date:** 20010821-20010923
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E.I. Conference No.: 58850 Proceedings of SPIE - The International Society for Optical Engineering (Proc SPIE Int Soc Opt Eng) (United States) 2001 4527/- (125-133)
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Document Type: Conference Paper; Conference Proceeding **Record Type:** Abstract
Language: English **Summary Language:** English
Number of References: 8

Colorado State University (CSU) operates the CSU CHILL radar at Greeley, Colorado as a NSF national facility. The VCHILL (Virtual-CHILL) project is aimed at developing and implementing protocols for **providing** the **raw** radar **data** to researchers in real-time over the internet. The Next Generation Internet has features, which could be used to transfer raw digitized radar signals (DRS) generated by the radar. A network transfer application using **TCP** to transfer DRS was developed. This network transfer application has been successfully tested to provide throughput of 90 Mbps over a 100 Mbps link and 300 Mbps over gigabit link. A software digital signal-processing module was developed. A network transfer application with **User Datagram protocol** as the underlying protocol was also developed. The performance of the software digital signal processing unit and the 'Reliable-DRS' version of the application to transfer the data across the network meet the requirements in most areas.

13/5/2 (Item 2 from file: 8)
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On TCP-friendly video transfer with consideration on application-level QoS
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Corresp. Author email: wakamiya@ics.es.osaka-u.ac.jp
Conference Title: 2000 IEEE International Conference on Multimedia and Expo (ICME 2000)
Conference Location: New York, NY United States **Conference Date:** 20000730-20000802
E.I. Conference No.: 58779 IEEE International Conference on Multi-Media and Expo (IEEE Int Conf Multi Media Expo) (United States) 2000 , IEEE 00TH8532 -/II/TUESDAY (843-846)
Publication Date: 20001201
Publisher: Institute of Electrical and Electronics Engineers Inc.
Document Type: Conference Paper; Conference Proceeding **Record Type:** Abstract
Language: English **Summary Language:** English
Number of References: 11

When both **TCP** and **UDP** sessions co-exist in the Internet, the performance of **TCP** sessions easily deteriorate because of congestion incurred by **UDP** sessions of real-time multimedia applications. In

this paper, we extend the **TCP**-friendly rate control protocol which originally pursuits the fair-share of link bandwidth among **TCP** and non-**TCP** sessions. With our proposed method, the achievable application-level QoS, such as perceived video quality and file transfer delay, becomes the same among **TCP** and non-**TCP** which traverse the same path. Through simulation experiments, we show that the **high quality video transfer** can be performed with our proposed method while satisfying the **TCP**-friendliness with regard to application-level QoS.

13/5/3 (Item 3 from file: 8)
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MPEG-TFRCP: Video transfer with TCP-friendly rate control protocol
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Conference Title: International Conference on Communications (ICC2001)
Conference Location: Helsinki Finland **Conference Date:** 20000611-20000614
Sponsor: IEEE; ICC GLOBECOM
E.I. Conference No.: 58417 IEEE International Conference on Communications (IEEE Int Conf Commun) (United States) 2001 , IEEE 1CH37240 1/- (137-141)
Publication Date: 20010913
Publisher: Institute of Electrical and Electronics Engineers Inc.
Document Type: Conference Paper; Conference Proceeding **Record Type:** Abstract
Language: English **Summary Language:** English
Number of References: 11

As the use of real-time multimedia applications increases, bandwidth available to **TCP** connections is oppressed by "greedy" **UDP** traffic and their performance extremely deteriorates. In order that both **TCP** and **UDP** sessions fairly co-exist in the Internet, **UDP** sessions should properly react against congestion as **TCP**. In this work, we implement a "**TCP**-friendly" rate control mechanism suitable to video applications and investigate its applicability to a real system through observation of the video quality at the receiver. It is shown through our experimental system that we can achieve **high-quality** and stable **video transfer** while fairly sharing the network bandwidth with **TCP** by applying our rate control at a control Interval of 16 or 32 times as long as RTT.

13/5/4 (Item 4 from file: 8)
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TCP-friendly video transfer
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Editor(s): Chiu, A.L.; Bucs, F.H.; Mei, R.D.
Editor(s) Affil.: AT and T Laboratories, United States
Conference Title: Internet Quality and Performance and Control of Network Systems
Conference Location: Boston, MA United States **Conference Date:** 20001106-20001107
Sponsor: SPIE
E.I. Conference No.: 58119 Proceedings of SPIE - The International Society for Optical Engineering (Proc SPIE Int Soc Opt Eng) (United States) 2001 4211/- (25-35)
Publication Date: 20010522
Publisher: SPIE
Item Identifier (DOI): 10.1117/12.417490
Document Type: Conference Paper; Conference Proceeding **Record Type:** Abstract
Language: English **Summary Language:** English
Number of References: 18

When both **TCP** and **UDP** connections co-exist in the Internet environment, the performance of **TCP** connections is heavily affected by the behavior of "greedy" **UDP** connections of real-time multimedia applications. In this paper, we propose a new **TCP**-friendly rate control protocol for video connections, called MPEG-TFRCP, to fairly share the link with **TCP** connections. To achieve fairness among **TCP** and **UDP** connections while performing **high quality video transmission**, we argue that (1) the interval of rate control must be appropriately determined, (2) the network condition must be accurately predicted, (3) the **TCP** throughput must be precisely estimated and (4) the video rate must be effectively adjusted. Although our algorithm is based on the existing proposals which do not satisfy all of those conditions, through careful considerations on the applicability of TFRCP to the actual video applications ours can achieve the **high- quality** MPEG-2 **video transfer** while satisfying the **TCP**-friendliness. Through simulation experiments, we show that the **TCP** throughput estimation based on pseudo-**TCP** feedback collection is acceptable and the rate adjustment based on the quantization control should be performed at the interval of 32 times as long as estimated RTT.

13/5/5 (Item 1 from file: 35)
DIALOG(R)File 35: Dissertation Abs Online
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Error concealment for robust image and video transmissions on the Internet
Author: Su, Xiao **Degree:** Ph.D.
Year: 2001
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Adviser: Benjamin J. Wah
Source: Volume 6208B of Dissertations Abstracts International.
PAGE 3693 . 190 PAGES

Coding and transmission of images and videos have been a popular but very challenging topic. Traditional coding algorithms are usually designed to optimize compression ratio in an error-free environment but not for the current Internet that is only a best-effort, packet-switched and unreliable network. This conflict presents a number of challenges for **high- quality** image and **video transmissions**.

Information loss and bandwidth limitation are two major factors that affect the quality of video streaming. In this thesis, we have developed various error concealment and reconstruction-based rate control schemes to address these two issues. First, we have proposed a sender-receiver based approach for designing a multiple-description video coder that facilitates recovery of <italic>packet losses</italic>. Second, we have studied artificial neural network-based reconstruction algorithms for compensating nonlinear <italic> compression losses</italic> due to multiple-description coding. Third, we have employed syntax-based packetization and decoder-side feedback for reducing <italic> propagation losses</italic>. Last, we have incorporated the reconstruction process in the design and evaluation of rate control schemes.

Likewise, delay and packet loss are two primary concerns for image transmissions. Existing approaches for delivering single-description coded images by **TCP** give superior quality but very long delays when the network is unreliable. To reduce delays, we have proposed to use **UDP** to deliver sender-receiver based multiple-description coded images. To further improve the trade-offs between delay and quality of either **TCP** delivery or **UDP** delivery, we have investigated a combined **TCP/UDP** delivery that can give shorter delay than pure **TCP** delivery and better quality than pure **UDP** delivery.

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DIALOG(R)File 2: INSPEC
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Towards high-quality video stream transmission for digital video production: a case study and experiences
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Book Title: International Conference on Computational Intelligence and Multimedia Applications 1998. ICCIMA 1998

Inclusive Page Numbers: 625-30
Publisher: World Scientific
Country of Publication: Singapore
Publication Date: 1998
Conference Title: Proceedings of International Conference on Computational Intelligence and Multimedia Applications
Conference Date: 9-11 Feb. 1998
Conference Location: Gippsland, Vic., Australia
Editor(s): Selvaraj, H. Verma, B.
Number of Pages: xix+893
Language: English
Document Type: Conference Paper (PA)

Experimental results on the issues of **high-quality video stream transmission** are presented. One of possible applications of **high-quality video** is for professional digital video production. From our study, we found that proper choices of packet size make the performance significantly different. Also, with **UDP** connection, transmission quality in terms of frame-loss rate could not be well satisfied even with enough computing power and network bandwidth. Disabling **UDP** check-sum operation in the experiments makes only very limited improvements for overall frame throughput. In addition, results from our experiments show that with some packet-loss recovery schemes like packet retransmission in **TCP**, video quality can be improved substantially with only a few additional system resources consumed. Meanwhile, large frame buffers may result in long delay, which is inappropriate for high-quality video production applications. (*5 refs.*)

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DIALOG(R)File 2: INSPEC
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Delivery of high quality uncompressed video over ATM to Windows NT desktop
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Country of Publication: USA
Publication Date: 1997
Conference Title: Proceedings of the USENIX Windows NT Workshop
Conference Date: 11-13 Aug. 1997
Conference Location: Seattle, WA, USA
Number of Pages: 150
Language: English
Document Type: Conference Paper (PA)

The emergence of high bandwidth applications such as medical visualization and virtual reality has exposed significant deficiencies in network, protocol, and end system design. We discuss important end system issues which arise when supporting applications demanding networked **delivery** and manipulation of **uncompressed video** to the desktop. Our experimental network environment consists of DEC Alpha workstations using the Windows NT 4.0 operating system and connected via an ATM switch. We present the design and initial results of a network architecture that demonstrates the creation, manipulation, and distribution of high quality uncompressed video using standard industry based technologies. In addition, we discuss networking performance results and present a simple Windows Sockets 2.0 cost model for **TCP**/IP and **UDP**/IP over ATM. An early potential market where this work is expected to have a direct impact is video editing in motion picture and television studios. In this context, we hope to provide cost effective networked solutions aimed at replacing costly dedicated video editing hardware with the versatile capabilities of general purpose workstations and nonproprietary network solutions. (*31 refs.*)

19/5/2 (Item 2 from file: 8)

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Performance evaluation of TCP extensions on ATM over high bandwidth delay product networks

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IEEE Communications Magazine (IEEE Commun Mag) 1999 37/7 (57-63)
Publication Date: 19990101
Publisher: IEEE
Item Identifier (DOI): 10.1109/35.774881
Document Type: Article; Journal **Record Type:** Abstract
Language: English **Summary Language:** English
Number of References: 18

Practical experiments in a satellite network environment assist in the design and understanding of future global networks. This article describes the practical experiences gained from TCP/IP on ATM networks over a high-speed satellite link and presents performance comparison studies of such networks with the same host/traffic configurations over local area and wide area networks. These comparison studies on the LAN, WAN, and satellite environments increase our understanding of the behavior of high-bandwidth networks. NASA's Advanced Communications Technology Satellite (ACTS), with its special characteristics and high data rate satellite channels, and the ACTS ATM Internetwork (AAI) were used in these experiments to deliver broad-band traffic. Network performance tests were carried out using application-level software (Netspec) on SONET OC-3 (155.52 Mb/s) satellite links. Finally, in this article we experimentally study the performance, efficiency, fairness, and aggressiveness of TCP Reno, TCP New Reno, and TCP SACK end hosts on ATM networks over high **BDP** networks.

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GPS Burst Detector W-Sensor

McCrady, D. D. ; Phipps, P.
Sandia National Labs., Albuquerque, NM.
Corporate Source Codes: 068123000; 9511100
Sponsor: Department of Energy, Washington, DC.
Report Number: SAND-94-2130C; CONF-9409185-1
1994 10p
Language: English Document Type: Conference proceeding
Journal Announcement: GRAI9508; ERA9508
Ion-GPS 94, Salt Lake City, UT (United States), 21 Sep 1994. Sponsored by Department of Energy, Washington, DC.
Order this product from NTIS by: phone at 1-800-553-NTIS (U.S. customers); (703)605-6000 (other countries); fax at (703)321-8547; and email at orders@ntis.fedworld.gov. NTIS is located at 5285 Port Royal Road, Springfield, VA, 22161, USA.
NTIS Prices: PC A02/MF A01
Country of Publication: United States
Contract Number: AC04-94AL85000

The NAVSTAR satellites have two missions: navigation and nuclear detonation detection. The main objective of this paper is to describe one of the key elements of the Nuclear Detonation Detection System (NDS), the Burst Detector W-Sensor (BDW) that was developed for the Air Force Space and Missle Systems Center, its mission on GPS Block IIR, and how it utilizes GPS timing signals to precisely locate nuclear detonations (NUDET). The paper will also cover the interface to the Burst Detector Processor (**BDP**) which links the BDW to the ground station where the BDW is controlled and where data from multiple satellites are processed to determine the location of the NUDET. The Block

IIR BDW is the culmination of a development program that has produced a state-of-the-art, space qualified digital receiver/processor that dissipates only 30 Watts, weighs 57 pounds, and has a 12in. (times) I4.2in. (times) 7.16in. footprint. The paper will highlight several of the key multilayer printed circuit cards without which the required power, weight, size, and radiation requirements could not have been met. In addition, key functions of the system software will be covered. The paper will be concluded with a discussion of the high speed digital signal processing and algorithm used to determine the time-of-arrival (TOA) of the electromagnetic pulse (EMP) from the NUDET.

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Experiences with MPEG-4 multimedia streaming
Shojania, H.; Li, B.
Conference Title: -ACM Multimedia 2001 Workshops- 2001 Multimedia Conference
Conference Location: Ottawa, Ont. Canada **Conference Date:** 20010930-20011005
Sponsor: ACM Special Interest Groups
E.I. Conference No.: 58703
Proceedings of the ACM International Multimedia Conference and Exhibition (Proc ACM Int Multimedia Conf Exhib) (United States) 2001 -/IV (492-494)
Publication Date: 20011027
Publisher: Association for Computing Machinery
Document Type: Conference Paper; Conference Proceeding **Record Type:** Abstract
Language: English **Summary Language:** English
Number of References: 2

With the advent of next-generation multimedia technologies such as very-low bit rate MPEG-4 codec, **multimedia** streaming of **high- quality video** and **audio** has become a near-term reality. The high compression ratio and error resilience offered by the MPEG-4 standard promise near-term popularity for rich contents and exceptional quality to consumers over affordable Internet connections, such as xDSL, cable modem and 3G wireless networks. Audio and video streaming applications are at the center of such scenarios; and Quality-of-Service (QoS) support in such applications is critical to their widespread acceptance. To the best of our knowledge, there has been no existing open-source MPEG-4 multimedia streaming applications in the academic community, which leads to the lack of research results using MPEG-4 streaming, especially with respect to Quality-of-Service support. In this work, we have implemented an open-source MPEG-4 multimedia streaming testbed in IP-based networks. In this paper, we show our experiences and lessons learned with such a testbed. First, we describe the algorithms and solutions used in our implemention testbed, emphasizing several critical issues. Second, through extensive experiments, we demonstrate measurements of bandwidth requirements and data loss for streaming a set of multimedia samples with different bit rates over **UDP**, which is ubiquitously available in the **TCP/IP** protocol stack on all consumer operating systems. Finally, future work for further improvements is also discussed.

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Experimental QoS performances of multimedia applications
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Corresp. Author/ Affil: Wang, Phil Yonghui: Columbia Univ, New York, United States
Conference Title: 19th Annual Joint Conference of the IEEE Computer and Communications Societies - IEEE INFOCOM2000: 'Reaching the Promised Land of Communications'
Conference Location: Tel Aviv, Isr **Conference Date:** 20000326-20000330
Sponsor: IEEE
E.I. Conference No.: 56703 Proceedings - IEEE INFOCOM (Proc IEEE INFOCOM) 2000 2/- (970-979)
Publication Date: 20000101
Publisher: IEEE

Document Type: Conference Paper; Conference Proceeding **Record Type:** Abstract
Language: English **Summary Language:** English
Number of References: 17

Several QoS provisioning mechanisms such as Differentiated Services (Diffserv) and Integrated Services (Intserv) have been recently devised and applied to bring Quality of Service (QoS) to the Internet. This paper studies end-end QoS performances of two QoS-demanding applications using different transport protocols. Both applications are tested in a real network environment, with end-end QoS provisioning by Intserv. They use QoSockets, a new extension of QoS specification and management to the Berkeley sockets. Their performances in terms of throughput, delay, jitter, and loss are measured under a number of test cases combining several factors: (1) single or multiple flows, with or without resource reservations; (2) normal, heavy, or overloaded scenarios; (3) uni- or bi- **directional streams**; and (4) **TCP** or **UDP** protocols. The experimental results show that the performances of two applications with the Intserv resource reservations are significantly improved, but not always guaranteed. It is also shown that **UDP** applications are able to get the requested QoS while **TCP** applications may not because of the nature of its bi-directional traffic flow. The paper provides detailed interpretation of the results and provides generic conclusions on application QoS.

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Efficient user-space protocol implementations with QoS guarantees using real-time upcalls
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IEEE/ACM Transactions on Networking (IEEE ACM Trans Networking) 1998 6/4 (374-388)
Publication Date: 19980101
Publisher: IEEE
Item Identifier (DOI): 10.1109/90.720871
Document Type: Article; Journal **Record Type:** Abstract
Language: English **Summary Language:** English
Number of References: 42

Two important requirements for protocol implementations to be able to provide quality of service (QoS) guarantees within the endsystem are: 1) efficient processor scheduling for application and protocol processing and 2) efficient mechanisms for data movement. Scheduling is needed to guarantee that the application and protocol tasks involved in processing each stream execute in a timely manner and obtain their required share of the CPU. We have designed and implemented an operating system (OS) mechanism called the real-time upcall (RTU) to provide such guarantees to applications. The RTU mechanism provides a simple real-time concurrency model and has minimal overheads for concurrency control and context switching compared to thread-based approaches. To demonstrate its efficacy, we have built RTU-based **transmission control protocol** (**TCP**) and **user datagram protocol** (**UDP**) protocol implementations that combine high efficiency with guaranteed performance. For efficient data movement, we have implemented a number of techniques such as: 1) **direct** movement of **data** between the application and the network adapter; 2) batching of input-output (I/O) operations to reduce context switches; and 3) header-data splitting at the receiver to keep bulk data page aligned. Our RTU-based user-space TCP/Internet protocol (TCP/IP) implementation provides bandwidth guarantees for bulk data connections even with real-time and `best-effort' load competing for CPU on the endsystem. Maximum achievable throughput is higher than the NetBSD kernel implementation due to efficient data movement. Sporadic and small messages with low delay requirements are also supported using reactive RTU's that are scheduled with very low delay. We believe that ours is the first solution that combines good data path performance with application-level bandwidth and delay guarantees for standard protocols and OS's.

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REAL TIME DATA ACQUISITION FOR A TIME PROJECTION CHAMBER USING A HIGH SPEED DEC-RT11 TO UNIX UDP-TCP/IP INTERFACE.

Thomas, Jim; Douglas, M.; Watanabe, R.; Henrikson, H.E.; Iqbal, M.Z.; Mitchell, L.W.; O'Callaghan, B.M.G.; Wong, H.T.-K.; Melvin, Jonathan D.
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Conference Title: Fifth Conf on Real-Time Comput Appl in Nucl, Part and Plasma Phys
Conference Location: San Francisco, CA, USA **Conference Date:** 19870512-19870514
IEEE Transactions on Nuclear Science (IEEE Trans Nucl Sci) 1987 NS-34/4 (845-848)
Publication Date: 19871201
Document Type: Article; Journal **Record Type:** Abstract
Language: English **Summary Language:** English
Number of References: 3

The authors describe the data acquisition system for a high-pressure xenon time projection chamber (TPC) that was built to study double beta decay. **Raw data** rates from the TPC exceed 200 kb/s. The TPC is operated through a CAMAC interface with a DEC LSI-11/73 computer networked to a Tektronix 6130 workstation. Data is transmitted at about 15 kb/s, although the network is capable of transmitting data at 80 kb/s. Although only one front-end computer and a single host are currently used on the system, several front-end computers and several hosts can be run simultaneously. This would have no effect on the overall transmission rate.

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Design of efficient and reliable TCP protocol service for power system communication applications

Author: Malik, Muhammed Tanveer **Degree:** D.Sc.
Year: 2001
Corporate Source/Institution: The George Washington University (0075)
Director: Robert Joseph Harrington
Source: Volume 6211B of Dissertations Abstracts International.
PAGE 5284 . 119 PAGES

The current standard based power system communication architectures are at their infancy stage right now. Most of the data retrieval ink are still legacy and the transport protocols don't support real time or any prioritized data flows. The future power system communication architecture requires real time transport protocols that can support multimedia and wireless applications. The multimedia based power system communication requirements are under progress, the standard based interfaces are being defined. The future utilities for inter and intra networks will be based on heterogeneous physical layer protocols, but upper layer transport protocols will be **TCP** and **UDP** and the associated services.

Many applications in power system communication will use **TCP** congestion control to regulate the transmission rate of a data packet stream. A natural way to achieve this goal is to transport the data packet stream on a TCP connection. However, because TCP implements both congestion and error control, transporting a **data** packet **stream directly** using a TCP connection forces the data packet stream to be subject to TCP's other properties caused by TCP error control, which may be inappropriate for these applications. The TCP out-of-band approach proposed in this thesis is a novel way of applying TCP congestion control to a data packet stream without actually transporting the data packet stream on a TCP connection. Instead, a TCP connection using the same network path as the data packet stream is set up separately and the transmission rate of the data packet stream is then associated with that of the TCP packets. Since the transmission rate of these TCP packets is under TCP congestion control, so is that of the data packet stream. Fore, since the data packet stream is not transported on a TCP connection. the regulated data packet stream is not subject to TCP error control. Because of this flexibility, the TCP out-of-band approach opens up many new opportunities, solves old problems, and improves the performance of some existing applications. All of these advantages will be demonstrated in the thesis. This thesis presents the power system communication requirements, current status of **TCP** and **UDP** performance issues and proposes new approach to design, implement,

and analyze the **TCP** out-of-band approach, and its successful applications in TCP trunking, wireless communication, add multimedia streaming, that will be heavily used in the modern power system communication architectures.

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Flow management for voice/ data transport over UDP/ TCP based networks
Author: Jeong, Seong-Ho **Degree:** Ph.D.
Year: 2000
Corporate Source/ Institution: Georgia Institute of Technology (0078)
Directors: John A. Copeland; Henry L. Owen
Source: Volume 6111B of Dissertations Abstracts International.
PAGE 6041 . 141 PAGES

An integrated voice/data network infrastructure may provide cost savings from both network and operations perspective, greater network use, and bandwidth flexibility. Therefore, service providers are interested in exploring the consequences of providing voice/data services over a single integrated switching and transport network based on Asynchronous Transfer Mode (ATM) or Internet Protocol (IP) technology. The two alternatives differ in how voice and data traffic is carried over them. The objective of this research is to present key issues concerning voice/data transport over ATM and IP networks and propose methods to solve the problems. ATM is well ahead of IP in addressing the quality of service (QoS) challenge, and it has been considered suitable for supporting the QoS requirements of voice traffic. One of the major issues concerning voice transport over ATM is the performance and survivability of voice-over-ATM systems. For IP, one of the biggest challenges will be to provide quality of service guarantees that are suitable for **high quality voice**. In IP networks, voice traffic is typically carried over user datagram protocol (UDP). To realize the performance requirements of UDP-based voice applications, it is necessary to provide appropriate bandwidth to the voice traffic so that the performance of voice traffic will not be seriously affected within the network during periods of congestion. Note that UDP flows do not typically back off when they encounter congestion, thus they are called unresponsive or aggressive flows. As a result, they aggressively use up more bandwidth than TCP friendly flows. This could eventually cause an Internet Meltdown. Therefore, while it is important to have router-based algorithms support **UDP** flows by assigning appropriate bandwidth, it is also necessary to protect responsive **TCP** flows from unresponsive or aggressive **UDP** flows so that all users can get a reasonable quality of service. In this research, we first analyzed the performance and survivability of voice-over-ATM systems in terms of trunking efficiency, QoS parameters, and network survivability parameters. We then proposed an architectural framework for flow management in IP networks. Based on the architectural framework, we also proposed and developed a set of router-based QoS mechanisms including queue policy, resource reservation, and metering. The proposed router-based QoS mechanisms provide a certain amount of bandwidth to **UDP** flows, protection of well-behaved **TCP** flows from unresponsive **UDP** flows, and bandwidth fairness between **TCP** flows.

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Experiences with multimedia applications over native ATM
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Language: English
Document Type: Journal Paper (JP)

Asynchronous transfer mode (ATM) is a high-speed networking technology that has gained wide acceptance for wide area and local area network environments. In the last few years, many applications have been deployed to run over ATM. However, most of these implementations use **TCP** or **UDP** as transport layers, with IP-over-ATM providing the network layer. Real desktop multimedia applications running over native ATM are yet to be deployed. In this paper, we present **raw data** performance results for **TCP-UDP/IP** and native ATM on Windows NT 4.0. We also describe our performance experiences with native ATM implementations of multimedia applications such as video conferencing and medical visualization. Finally, we demonstrate the benefit of native ATM over TCP/IP on quality of service (QoS) parameters such as jitter, in cases when multiple multimedia applications run concurrently. It is hoped that the lessons and experiences gained will be useful to designers and implementers of native ATM services on popular operating system platforms such as Windows NT 4.0. (*26 refs.*)

23/5/11 (Item 2 from file: 266)
DIALOG(R)File 266: FEDRIP
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Identifying No.: 9980637 **Agency Code:** NSF
Traffic Management in IP Networks
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Performing Org.: Ohio State University Research Foundation, Computer & Info Sci. , Columbus , OH 43210-1277
Project Monitor: Sollins, Karen R.
Sponsoring Org.: National Science Foundation, ANI , 4201 Wilson Boulevard , Arlington , Virginia 22230
Dates: 20001001 **To** 20010930 **Fy :** 2000 **Funds:** $241,064 (200000)

There is an immense demand for quality of service (QoS) in the Internet. One key element of quality of service is traffic management. Since the network traffic is bursty, it is difficult to make any QoS guarantees without proper control of traffic. Currently, Internet Protocol (IP) has only minimal traffic management capabilities. The packets are dropped when the queue exceeds the buffer capacity. The transmission control protocol (TCP) uses the packet drop as a signal of congestion and reduces its load. While in the past, this strategy has worked satisfactorily, there is need for better strategies for two reasons. First, a large part of the traffic, particularly, voice and video traffic does not use **TCP**. Continuous media traffic uses **User Datagram Protocol** (**UDP**). The proportion of **UDP** traffic is increasing at a faster pace than **TCP** traffic. The **UDP** traffic is congestion insensitive in the sense that **UDP** sources do not reduce their load in response to congestion. Second, the bandwidth of the networks as well as the distances are increasing. For very high distance-bandwidth product networks, packet drop is not the optimal congestion indication. Several megabytes of data may be lost in the time required to detect and respond to packet losses. Therefore, a better strategy for traffic management in IP networks is required. Recognizing the need for **direct** feedback of congestion **information**, the Internet Engineering Task Force (IETF) has come up with an Explicit Congestion Notification (ECN) method for IP routers. A bit in the IP header is set when the routers are congested. ECN is much more powerful than the simple packet drop indication used by existing routers and is suitable for high distance-bandwidth networks. Unfortunately, to realize the full potential of ECN, several questions need to be answered. In this research proposal, the PIs propose a comprehensive program of research on traffic management in IP networks. They propose to develop a new set of traffic management algorithms for IP networks based on Explicit Congestion Notification mechanism. A total of 18 different issues will be analyzed. The PIs have identified potential solutions and approaches for each of these issues. Specifically, they propose to work on a new congestion detection and buffer management scheme for routers, a mechanism for TCP to react to ECN messages from the network. One of the important goals of this research is to make TCP traffic management algorithm free of any bias based on round trip time and number of congested gateways traveled. The proposed research will be based on theoretical analysis and simulations. The PI's approach will be a formal analysis of simple scenarios, heuristic analysis of more complex scenarios and validation using simulations. The emphasis will be to develop simple solutions. However, the performance lost in exchange of simplicity will be theoretically analyzed. Traffic management is the key in providing QoS. Currently, a significant amount of NSF, DARPA, and other research funding as well as energy in networking is being spent on

QoS issues. When QoS based solutions (integrated services, differentiated services, or multiprotocol label switching) are deployed, the need for traffic management will become apparent and the PIs expect to see an immediate need for proper methods for traffic management. This proposal is, therefore, timely and important.

13/3,K/1 (Item 1 from file: 275)
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SIP pundits voice support - Nascent protocol promises integrated VoIP, IM, despite challenges.(Session Initiation Protocol pushed as standard for packet-based voice)
Shafer, Scott Tyler
InfoWorld , 24 , 32 , 25
August 12 , 2002
Language: English **Record Type:** Fulltext
Word Count: 1048 **Line Count:** 00087

instant messaging application that is garnering the most interest from vendors and analysts.
Support for basic text messaging will be boosted by the capability of **transmitting** video and even **voice** communications **directly** between two users. SIP will likely be the mechanism that enables this type of connection, effectively replacing H.323, a current standard also used to...
...a lot like HTML, thus making them easy to read.
"SIP is flexible and simple and able to run on top of other protocols like **TCP** and **UDP** (**User Datagram Protocol**)," said Dave Passmore, an analyst at Burton Group during its Catalyst Conference in San Francisco.
In addition SIP uses e-mail style addressing and uses...

13/3,K/2 (Item 2 from file: 275)
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Small Device Remote Access.(handheld, other devices)(Technology Information)
Angel, Jonathan
Network , NA
Nov 1 , 1999
Language: English **Record Type:** Fulltext; Abstract
Word Count: 9500 **Line Count:** 00786

Computing Magazine, at www.pencomputing. com/palm/, has news and information about both CE and Palm platforms. It also maintains an AvantGo server that can **deliver** its **content directly** to Palm users.
Puma Technology offers a white paper on synchronization at www.pumatech.com/ syncwp.html.
The Wireless Application Protocol (WAP) Forum is an...
...1.2 of its specification, which describes both communications protocols and an application development environment.
WAP protocols build on Internet standards such as IP and **UDP**. However, WAP eschews HTTP and **TCP** (considered unsuitable for the long latencies and limited bandwidth in wireless networks) in favor of its own method of compressed binary transmission. WAP sessions cope...

13/3,K/3 (Item 3 from file: 275)
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WavePhore Inks Satellite Network Deal With Telenor.
Newsbytes , pNEW10310038
Oct 31 , 1997

Language: English **Record Type:** Fulltext
Word Count: 517 **Line Count:** 00047

StatMux and Time Division Multiplexor systems, to provide further efficiency and flexibility in transmitting various data, Calder said. The network can also be upgraded to **deliver data directly** to a customer LAN via an Ethernet port using **TCP/IP** or **UDP**.
Telenor selected WavePhore Networks as the data network equipment supplier for its new SCPC satellite system because of the company's experience working with other...

13/3,K/6 (Item 2 from file: 621)
DIALOG(R)File 621: Gale Group New Prod.Annou.(R)
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Voxware Speech and Audio Compression Technologies Included With NetShow Services
PR Newswire , p 708NYW034
July 8 , 1998
Language: English **Record Type:** Fulltext
Article Type: Article
Document Type: Newswire ; Trade
Word Count: 744

universal player that plays most local and streamed media file types. The NetShow Services enable Internet Providers, Web-site hosts and corporations to develop and **deliver high quality audio**, **video** and mixed multimedia over the Internet or the intranet. The NetShow Services supports both live and on-demand services using the **TCP**, **UDP** and HTTP protocols.
Voxware's speech codec family supports high quality, real-time speech at bit rates of 2.4 kbps and 2.9kbps. Optimized...

13/3,K/9 (Item 1 from file: 636)
DIALOG(R)File 636: Gale Group Newsletter DB(TM)
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WavePhore Inks Satellite Network Deal With Telenor 10/31/97
Pietrucha, Bill
Newsbytes , p N/A
Oct 31 , 1997
Language: English **Record Type:** Fulltext
Document Type: Newswire ; General Trade
Word Count: 493

StatMux and Time Division Multiplexor systems, to provide further efficiency and flexibility in transmitting various data, Calder said. The network can also be upgraded to **deliver data directly** to a customer LAN via an Ethernet port using **TCP/IP** or **UDP**.
Telenor selected WavePhore Networks as the data network equipment supplier for its new SCPC satellite system because of the company's experience working with other...

13/3,K/12 (Item 2 from file: 16)
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Secure IP Access At Last With Virtual TCP Online
Network Computing , p 52
Sept 15 , 1996
Language: English **Record Type:** Fulltext
Document Type: Magazine/Journal ; Trade
Word Count: 537

on their WinSock PCs can access their company's network and applications without fear of the connection being monitored or spoofed.

A Safe Haven Virtual **TCP** Online achieves this by capturing and encrypting all **TCP** and **User Datagram Protocol** (**UDP**) packets generated on the PC, and then **sending** the encrypted **data directly** to the proxy host running on the internal corporate network. The proxy host decrypts the packets and sends them to their original destinations.

When data...

13/3,K/13 (Item 1 from file: 148)
DIALOG(R)File 148: Gale Group Trade & Industry DB
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QOS: what can service providers deliver?(includes related article on vendor activity and glossary)(quality of service at public packet networks)
Wexler, Joanie
Business Communications Review , 29 , 4 , 25(6)
April , 1999
Language: English **Record Type:** Fulltext; Abstract
Word Count: 4839 **Line Count:** 00404

both the customer's and the carrier's networks: shaping, marking and prioritizing traffic. This is done through a variety of mechanisms, including mapping a **TCP** (or **UDP**) port - which is bound to a particular application - to a certain class of service. But mechanisms on both sides of the access link - in CPE...

...ability to regulate traffic among the PVCs (as in MCI's Priority PVC offering), this really does no good," said John Scarborough, MCI WorldCom's **director** of virtual **data** services.

In the near term, service **providers** are beginning to offer IP QOS services based on traffic shaping. Last November worldwide ISP UUNet (www.uu.net) announced Access Optimization Services (AOS). The...

13/3,K/18 (Item 4 from file: 15)
DIALOG(R)File 15: ABI/Inform(R)
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Streaming audio and video: Bringing the intranet to life
Gibbs, Mark
Network World v13n49 pp: S11-S12
Dec 2, 1996
Word Count: 1136

added to the streaming system to make it easier to provide advanced features such as fast forward and jump. A streaming server also can theoretically **provide** better service for multiple clients than **direct data** retrieval through a Web server. The connection between the client and server is important. Server-based streaming products (for example, RealAudio and StreamWorks) use **User Datagram Protocol** (**UDP**), the **TCP**/IP protocol for sending streamed data. Because of this, these products potentially provide better throughput than those that don't use a server. Without a server, the data is retrieved via an HTTP connection using **TCP**. Due to the overhead of **TCP**'s error-correction mechanism, the characteristics of HTTP communications reduce the data rate significantly. That said, **UDP**, because it doesn't provide error correction, can degrade performance and rendering quality. However, most of the current audio products handle quite high numbers of...